CLAIMS

1. A method for modelling, i.a. analyzing and/or synthesizing, a windowed signal such as sound or speech signals, by computing the frequencies and complex amplitudes from the signal using a nonlinear least squares method, whereby the computational complexity is reduced by taking into account the bandlimited property of the window.

2. A method according to claim 1 using a stationary nonharmonic signal model according to Eq. (2) and/or a harmonic signal model according to Eq. (3) and/or a nonstationary nonharmonic model according to Eq. (4).

3. A method according to claim 1 or 2, comprising the computation of the spectrum as a linear combination of the frequency responses of the window according to Eq. (11) for the stationary nonharmonic model, Eq. (12) of the harmonic model and Eq. (13) for the nonstationary model, whereby only the main lobes of the responses are computed by using look-up tables.

4. A method according any of claim 1-3, comprising a pre-processing step which comprises: sorting the frequencies, eliminating frequencies which are close to one another and determining the number of relevant diagonal bands D.

5. A method according any of claim 1-4, comprising the step of computing the stationary complex amplitudes, preferably by solving the equations given in Eq. (19), using Eq. (20) such that only the elements around the diagonal of $\mathbf{B}$ are taken into account, whereby a shifted form $\overleftarrow{\mathbf{B}}$ is computed containing only D diagonal bands of $\mathbf{B}$ according to Eq. (27) and Eq. (20), whereby the computation of the Eq. (20) requires the computation of the frequency response of the window and the square window denoted by $W(m)$ and $Y(m)$ respectively, and solving equation given by Eq. (19) directly from $\overleftarrow{\mathbf{B}}$ and $\mathbf{C}$ (Eq. (28)) by an adapted gaussian elimination procedure.

6. A method according to any of claim 1-5, further comprising the step of optimizing the frequencies for the stationary nonharmonic model preferably by solving the equation given in Eq. (34), using Eq. (42) such that only elements around the diagonal of $\mathbf{H}$ are taken into account, whereby a shifted form $\overleftarrow{\mathbf{H}}$ is computed containing only D diagonal bands according to Eq. (36) and Eq. (42), whereby the the gradient h

is computed from the residual spectrum R_m, amplitude A_l and frequencies ω_l, and requires the computation of derivative of the frequency response of the window $W'(m)$, whereby the first term of H requires the computation of the second derivative of the frequency response of the square window denoted $Y''(m)$, whereby the second term of H is computed from the residual spectrum R_m, amplitude A_l and frequencies ω_l, and requires the computation of the second derivative of the frequency response $W''(m)$, whereby the parameter λ_1 allows to switch between different optimization methods and the parameter λ_2 regularizes the system matrix, and computing the optimization step by solving the the system of equations directly on $\overleftarrow{\mathrm{H}}$ and $\mathbf{h}$ according to Eq. (37) by an adapted gaussian elimination procedure.

7. A method according to any of claim 1-6, further comprising the step of optimization the frequencies for the harmonic signal model, preferably by computing the optimization step solving Eq. (48) using Eq. (49), whereby the gradient h is computed from the residual spectrum R_m, amplitude A_l and frequencies ω_l, and requires the computation of derivative of the frequency response of the window $W'(m)$, whereby the first term of H requires the computation of the second derivative of the frequency response of the square window denoted $Y''(m)$, whereby the second term of H is computed from the residual spectrum R_m, amplitude A_l and frequencies ω_l, and requires the computation of the second derivative of the frequency response $W''(m)$, whereby the parameter λ_1 allows to switch between different optimization methods and the parameter λ_2 regularizes the system matrix.

8. A method according to any of claim 1-7, further comprising the step of computing the polynomial complex amplitudes preferably by solving the equation given in Eq. (55), using Eq. (63) such that only the elements around the diagonal of B are taken into account, whereby a shifted form $\overleftarrow{\mathrm{B}}$ is computed containing only PD diagonal bands of B according to Eq. (64) and Eq. (63), whereby the computation is required of the frequency response of the square window and its derivatives $\frac{\partial^p}{\partial m^p}Y(m)$, whereby the computation is required of the frequency response of the window and its derivatives $\frac{\partial^p}{\partial m^p}W(m)$, and solving the equation given by Eq. (55) directly from $\overleftarrow{\mathrm{B}}$ and C by an adapted gaussian elimination procedure.

9. A method according to any of claim 1-8, further comprising the step of computing

instantaneous frequencies and instantaneous amplitudes according to Eq. (69), whereby the instantaneous frequency can be used as a frequency estimate for the next iteration as expressed in Eq. (73).

10. A method according to any of claim 1-9, further comprising the step of computing damping factor according to Eq. (78), in case that the amplitudes are exponentially damped.

11. A method to compute the frequency response of a window with length M zero padded up to a length N by using a scaled table look-up according to Eq. (82).

12. Apparatus wherein the method according to any of the previous claims 1 to 10 is implemented.

13. Use of a method according to any of the claims 1 to 10 or an apparatus according to claim 11 for accurate pitch estimation.

14. Use of a method according to any of the claims 1 to 10 or an apparatus according to claim 11 for arbitrary sample rate conversion.

15. Use of a method according to any of the claims 1 to 10 or an apparatus according to claim 11 for parametric/sinusoidal audio coders, where the noise residual, amplitudes and frequencies are encoded in a bitstream which is stored, broadcasted or transmitted at the sender side, the receiver decodes the bitstream back to the parameters and synthesizes the sound.

16. Use of a method according to any of the claims 1 to 10 or an apparatus according to claim 11 for audio effects whereby the noise r_n, the amplitudes $\bar{A}$ and frequencies $\bar{\omega}$ are manipulated by an effects processor yielding r_n^*, $\bar{A}^*$ and $\bar{\omega}^*$ and synthesized with with these modified parameters.

17. Use of a method according to any of the claims 1 to 10 or an apparatus according to claim 11 for source separation, whereby sinusoidal components originating from the same sound source are grouped and synthesized separately.

18. Use of a method according to any of the claims 1 to 10 or an apparatus according to claim 11 for automated annotation and transcription whereby the the signal is segmented according to the values of the amplitudes and frequencies.